DETAILED ACTION

Claim Rejections - 35 USC § 101

1. 35 U.S.C. 101 reads as follows:

> Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

2. Claims 1-13 are rejected under 35 U.S.C. 101 because the claimed invention is directed to non-statutory subject matter.

Claims 1-13 are directed to a process (i.e. a time-scale modification method). A process claim, to be statutory under § 101, must pass the machine-or-transformation test (M-or-T test), which ensures that the process is limited to a particular practical application. In accordance with the M-or-T test, the claimed process must:

(1) be tied to a particular machine or apparatus (machine implemented); or

(2) particularly transform a particular article to a different state or thing.

Additionally, the particular machine tie or particular transformation must meet two corollaries to pass the test for subject matter eligibility. First, the use of the particular machine or transformation of the particular article must impose a meaningful limit on the claim's scope. Second, the use of the particular machine or the transformation of the particular article must involve more than insignificant "extra-solution" activity.

With respect to claim 1, there is no explicit recitation of a particular machine or apparatus implement any steps of the method. Furthermore, it appears that none of the recited steps would inherently require a particular machine or apparatus. That is, the steps of defining a number of samples, calculating a shift value, defining a second number of samples, and accumulating an error could conceivably be performed by a

human manually performing such calculations. Since all steps of the method could be performed manually by a human, claims 1-13 are not properly tied to a particular machine or apparatus, and thus fail test (1), above.

Additionally, there is no particular transformation involved in the claims. The method is directed to time-scale modification of a digital signal, i.e. a mere manipulation of electronic data. There is no particular article claimed that is transformed into a different state or thing, thus claims 1-13 fail test (2), above.

Since claims 1-13 fail both tests (1) and (2) above, the claims do not qualify as statutory processes under 35 U.S.C. 101.

Claim Rejections - 35 USC § 112

3. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

4. Claim 10 rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Claim 10 recites the limitations "the upper and lower limit" and “the allowed error range" in line 2 of the claim. There is insufficient antecedent basis for these limitations in the claim.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

6. Claims 7 and 10 are rejected under 35 U.S.C. 102(b) as being anticipated by Ware (U.S. Patent 5,664,044).

In regard to claim 7, Ware discloses a time-scale modification method for a digital audio/video signal (Fig. 1), in which an input digital audio/video signal is separated into an audio signal and a video signal, each of which is time-scaled with a same time-scale α (audio frames and video frames are placed into separate buffers, and the audio time scaling factor C is set equal to the video time scaling factor RV, column 6, lines 6-10), the method comprising steps of:

a) calculating periodically a real time-scale of a time-scaled video signal obtained by time-scaling the video signal based on the time-scale α (step 104, a playback time T based on the scaling factor C is determined, column 6, lines 10-21);

b) determining whether a real time-scale of a current period of the time-scaled video signal differs from that of a previous period (step 109, a determination is made if the audio playback time is not equal to T, column 6, lines 54-60), wherein, if different, the real time-scale of the current period is provided as a target time-scale α', the target time-scale α' becoming a reference for the time-scale modification of the audio signal

(step 110, if the audio playback time is not equal to T, the audio time scale C is adjusted, column 6, lines 54-60) ; and

c) segmenting a sample stream of the input audio signal into a plurality of overlapping analysis windows, changing the length of the overlapping area into a length corresponding to the target time-scale α', and weighted-synthesizing the overlapping area, thereby modifying into a time-scaled output audio signal (the audio time scale C is used in a time domain harmonic scaling process on the audio signal, column 6, lines 54-60; a time domain harmonic scaling process segments a sample stream of audio data with overlapping windows corresponding to a target time scale (C), and weight-synthesizes the overlapping areas, see column 1, lines 37-52).

In regard to claim 10, Ware discloses the upper and lower limit of the allowed error range is determined within an error range such that an unsynchronization between the audio and video signals is not recognized during their time-scaled reproduction (an audio buffer is monitored for overflow or underflow so that that audio and video signals remain synchronized, column 3, lines 29-35).

Claim Rejections - 35 USC § 103

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

8. Claims 14, 15, and 22-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hejna (U.S. Patent 6,598,228), in view of Ware.

In regard to claim 14, Hejna discloses a method of reproducing a broadcast signal using an apparatus, which receives a transport stream of a digital television broadcast signal compressed and coded in a MPEG mode and reproduces video and audio signals in real-time (live broadcasts, column 19, lines 2-6), the method comprising steps of:

a) storing sequentially a digital television broadcast signal being received in a storage means at least after a user inputs a phone-break key (a user presses a “pause and record button”, which initiates recording of the live broadcast, column 19, lines 6-14) ;

b) after the user presses a return key, reading the stored broadcast signal in a FIFO mode and time-scaling the respective retrieved video and audio signals with a predetermine time-scale (when the user resumes playback, the broadcast is played back from the pause position and time scaled at a rate that allows the recorded signal to "catch up" with the live signal, column 19, lines 14-30); and

c) outputting the time-scaled video and audio signals in place of a broadcast signal being currently received (the recorded signal is output until the user is "caught up" with the live broadcast, column 19, lines 14-30).

Although Hejna discloses time scaling of the retrieved audio and video signals, Hejna does not specifically disclose the time-scaling of the audio signal is performed based on a real time-scale α of the produced video signal, the real time-scale of the

video signal obtained by the time-scaling of the video signal being calculated by applying the predetermine time-scale, an audio sample stream of an input signal is segmented into a plurality of overlapping analysis windows, the length of the overlapping area is changed into a length corresponding to the real time-scale α of the video signal, and the overlapping area is weighted-synthesized, thereby converting into a time-scaled output signal.

Ware discloses a method for time scaling a retrieved audio and video signal where the time-scaling of the audio signal is performed based on a real time-scale α of the produced video signal, the real time-scale of the video signal obtained by the time-scaling of the video signal being calculated by applying the predetermine time-scale (step 104, a playback time T based on the scaling factor C is determined, column 6, lines 10-21), an audio sample stream of an input signal is segmented into a plurality of overlapping analysis windows, the length of the overlapping area is changed into a length corresponding to the real time-scale α of the video signal, and the overlapping area is weighted-synthesized, thereby converting into a time-scaled output signal (the audio time scale C is used in a time domain harmonic scaling process on the audio signal, column 6, lines 54-60; a time domain harmonic scaling process segments a sample stream of audio data with overlapping windows corresponding to a target time scale (C), and weight-synthesizes the overlapping areas, see column 1, lines 37-52).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify Hejna to use the audio and video time scaling technique disclosed by Ware, because this would ensure that the audio and video signals would remain

synchronized during the time-scaled “catch up” period, as suggested by Ware (column 1, lines 28-35).

In regard to claim 15, Hejna discloses outputting a broadcast signal being currently received instead of the stored broadcast signal, if a time difference between a broadcast signal reproduced by applying the time-scale α as a value for a high speed reproduction mode and the broadcast signal being currently received falls within a certain desired error range (see Fig. 9C, at a time Tr to Tv, the broadcast signal is reproduced by applying accelerated playback. When the reproduced broadcast signal is “caught up“with the live broadcast signal at time TV, the live broadcast signal is output, column 19, lines 35-62).

In regard to claim 22, Hejna does not disclose a step of uncompressing and decoding the video and audio signals respectively by means of a MPEG decoder before time-scaling the broadcast signal stored in the storage means.

Ware discloses a step of uncompressing and decoding the video and audio signals respectively by means of a MPEG decoder before time-scaling the broadcast signal stored in the storage means (input MPEG data is decoded by audio and video decoders prior to time scaling by the time domain harmonic scalar, see Fig. 4 and column 4, lines 44-67).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify Hejna to uncompressing and decoding the video and audio signals

respectively by means of a MPEG decoder, because MPEG is a widely know and used video and audio coding standard and using an MPEG decoder would insure greater interoperability.

In regard to claim 23, Hejna discloses the time-scaling of the video signal is performed by an adjustment of the output time interval of the video frames so as to be as fast as the time-scale, or a reduction of the number of output frames so as to be as low as the time-scale, or a combination of the above two (frames are skipped, column 7, lines 26-37).

In regard to claim 24, Hejna does not disclose the adjustment of the output time interval of the video frames is carried out an adjustment of the value of presentation time stamp of the video frame.

Ware discloses the adjustment of the output time interval of the video frames is carried out an adjustment of the value of presentation time stamp of the video frame (a video frame's presentation time T is adjusted to meet the time scaling time C, column 6, lines 6-18).

It would have been obvious to one of ordinary skill in the art at the time of invention to adjust the output time interval of the video frame based on an adjustment of the value of presentation time stamp of the video frame, because this would ensure synchronization with the audio data.

9. Claim 16 is rejected under 35 U.S.C. 103(a) as being unpatentable over Hejna, in view of Ware, and further in view of Barton (U.S. Patent 6,327,418).

In regard to claim 16, Hejna and Ware do not disclose when the phone-break period between the phone-break key input and the return key input exceeds the maximum storage time of the storage means, replacing with the broadcast signal being currently received the stored broadcast signal, in sequence from an earlier stored one, and changing the start address of the phone-break period from the current time into an address of a broadcast signal stored before the maximum storing time.

Barton discloses a method for providing a phone-break period (pausing), wherein when the phone-break period between the phone-break key input and the return key input exceeds the maximum storage time of the storage means, replacing with the broadcast signal being currently received the stored broadcast signal, in sequence from an earlier stored one, and changing the start address of the phone-break period from the current time into an address of a broadcast signal stored before the maximum storing time (when the pause is longer than the length of a cache, the pause indicator is unlocked and pointed at the earliest available block in the cache, column 8, line 56 to column 9, line 21).

It would have been obvious to one of ordinary skill in the art at the time of invention to further modify Hejna and Ware to replace with the broadcast signal being currently received the stored broadcast signal, in sequence from an earlier stored one, and change the start address of the phone-break period from the current time into an address of a broadcast signal stored before the maximum storing time, because this

would ensure the user would retain as much of the broadcast signal after the pause point as possible.

10. Claims 17 and 19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Rakib (U.S. Patent 6,970,127), in view of Ware.

In regard to claim 17, Rakib discloses a method of reproducing a broadcast signal using an apparatus, which receives a transport stream of a digital television broadcast signal compressed and coded in a MPEG mode and reproduces video and audio signals in real-time, the method comprising steps of:

a) storing sequentially the broadcast signal in a storage means (a hard disk records broadcast video and audio data, column 11, lines 45-49) ;

b) when a user's back-and-slow key input is detected, reading the stored broadcast signal in a FIFO mode, starting from a broadcast signal received before a certain period of time from that time point, and time-scaling the respective retrieved video and audio signals with a predetermine time-scale so as to enable a low speed reproduction (an instant replay function jumps the program back 8 seconds and replays the audio and video signals in slow motion, column 12, lines 39-47); and

c) outputting the time-scaled video and audio signals in place of a broadcast signal being currently received (the slow motion video is replayed instead of the broadcast signal, column 12, lines 39-47).

While Rakib discloses enabling a low speed reproduction, Rakib does not specifically disclose the time-scaling of the audio signal is performed based on a real

time-scale α of the produced video signal, the real time-scale of the video signal obtained by the time-scaling of the video signal being calculated by applying the predetermine time-scale, an audio sample stream of an input signal is segmented into a plurality of overlapping analysis windows, the length of the overlapping area is changed into a length corresponding to the real time-scale α of the video signal, and the overlapping area is weighted-synthesized, thereby converting into a time-scaled output signal.

Ware discloses a method for time scaling a retrieved audio and video signal where the time-scaling of the audio signal is performed based on a real time-scale α of the produced video signal, the real time-scale of the video signal obtained by the time-scaling of the video signal being calculated by applying the predetermine time-scale (step 104, a playback time T based on the scaling factor C is determined, column 6, lines 10-21), an audio sample stream of an input signal is segmented into a plurality of overlapping analysis windows, the length of the overlapping area is changed into a length corresponding to the real time-scale α of the video signal, and the overlapping area is weighted-synthesized, thereby converting into a time-scaled output signal (the audio time scale C is used in a time domain harmonic scaling process on the audio signal, column 6, lines 54-60; a time domain harmonic scaling process segments a sample stream of audio data with overlapping windows corresponding to a target time scale (C), and weight-synthesizes the overlapping areas, see column 1, lines 37-52).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify Rakib to use the audio and video time scaling technique disclosed

by Ware, because this would ensure that the audio and video signals would remain synchronized during the low speed reproduction, as suggested by Ware (column 1, lines 28-35).

In regard to claim 19, Rakib discloses a method of reproducing a broadcast signal using an apparatus, which receives a transport stream of a digital television broadcast signal compressed and coded in a MPEG mode and reproduces video and audio signals in real-time, the method comprising steps of:

a) storing sequentially the broadcast signal in a storage means at least after a user inputs an immediate-slow key (a hard disk records broadcast video and audio data, column 11, lines 45-49);

b) reading the stored broadcast signal in a FIFO mode starting from the point of inputting the immediate-slow key and time-scaling the respective retrieved video and audio signals with a predetermine time-scale so as to enable a low speed reproduction (live TV programs are played in slow motion, column 12, lines 39-47), and

c) outputting the time-scaled video and audio signals in place of a broadcast signal being currently received (the slow motion video is played instead of the broadcast signal, column 12, lines 39-47).

While Rakib discloses enabling a low speed reproduction, Rakib does not specifically disclose the time-scaling of the audio signal is performed based on a real time-scale α of the produced video signal, the real time-scale of the video signal obtained by the time-scaling of the video signal being calculated by applying the

predetermine time-scale, an audio sample stream of an input signal is segmented into a plurality of overlapping analysis windows, the length of the overlapping area is changed into a length corresponding to the real time-scale α of the video signal, and the overlapping area is weighted-synthesized, thereby converting into a time-scaled output signal.

Ware discloses a method for time scaling a retrieved audio and video signal where the time-scaling of the audio signal is performed based on a real time-scale α of the produced video signal, the real time-scale of the video signal obtained by the time-scaling of the video signal being calculated by applying the predetermine time-scale (step 104, a playback time T based on the scaling factor C is determined, column 6, lines 10-21), an audio sample stream of an input signal is segmented into a plurality of overlapping analysis windows, the length of the overlapping area is changed into a length corresponding to the real time-scale α of the video signal, and the overlapping area is weighted-synthesized, thereby converting into a time-scaled output signal (the audio time scale C is used in a time domain harmonic scaling process on the audio signal, column 6, lines 54-60; a time domain harmonic scaling process segments a sample stream of audio data with overlapping windows corresponding to a target time scale (C), and weight-synthesizes the overlapping areas, see column 1, lines 37-52).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify Rakib to use the audio and video time scaling technique disclosed by Ware, because this would ensure that the audio and video signals would remain

synchronized during the low speed reproduction, as suggested by Ware (column 1, lines 28-35).

11. Claims 18 and 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Rakib, in view of Ware, and further in view of Hejna.

In regard to claims 18 and 20, Rakib and Ware do not disclose a) when the user inputs a return key, time-scaling the stored broadcast signal for a high speed reproduction by modifying the time-scale into a value for a high speed reproduction mode, and b) outputting a broadcast signal being currently received instead of the stored broadcast signal, if a time difference between a broadcast signal being reproduced in a high speed mode and the broadcast signal being currently received falls within a certain desired error range.

Hejna discloses a time scaling method wherein a) when the user inputs a return key, time-scaling the stored broadcast signal for a high speed reproduction by modifying the time-scale into a value for a high speed reproduction mode (when the user resumes playback, the broadcast is played back and time scaled at a rate that allows the recorded signal to "catch up" with the live signal, column 19, lines 14-30), and b) outputting a broadcast signal being currently received instead of the stored broadcast signal, if a time difference between a broadcast signal being reproduced in a high speed mode and the broadcast signal being currently received falls within a certain desired error range (see Fig. 9C, at a time Tr to Tv, the broadcast signal is reproduced by applying accelerated playback. When the reproduced broadcast signal is “caught

up“with the live broadcast signal at time TV, the live broadcast signal is output, column 19, lines 35-62).

It would have been obvious to one of ordinary skill in the art at the time of invention to further modify Rakib and Ware to time scale the stored broadcast signal in a high speed reproduction mode until the time difference between a broadcast signal and the reproduction signal were within a certain error range, because this would allow a user to catch up to the broadcast signal without deleting or skipping some portion of the reproduced broadcast signal, as suggested by Hejna (column 19, lines 20-30).

Allowable Subject Matter

12. Claim 21 is objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

The following is a statement of reasons for the indication of allowable subject matter:

Rakib, Ware, Hejna, and Barton do not disclose or suggest in addition to the other features of the claim, time scaling by determining a modified analysis interval and a compensated analysis interval, and accumulating an error between a real production time and a computed reproduction time as claimed.

Conclusion

13. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Logan et al. (U.S. Patent 5,371,551) disclose an additional broadcast playback device with time scaling. Tillman et al. (U.S. Patent 6,496,980) disclose a system with a playback in slow motion button. Florencio (U.S. Patent 7,337,108) and Wilson (U.S. Patent 5,842,172) disclose additional audio time scale modification techniques.

14. Any inquiry concerning this communication or earlier communications from the examiner should be directed to BRIAN L. ALBERTALLI whose telephone number is (571)272-7616. The examiner can normally be reached on Monday-Thursday, 8 AM to 6:30 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, David Hudspeth can be reached on (571) 272-7843. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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/Brian L Albertalli/
Examiner, Art Unit 2626